CLAIMS

1.   A multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames, each said input data frame including subband data and their scale factors that have been compressed and packed;

5   A human input device (HID) for receiving input from a user;

An application programming interface (API) that generates a list of audio components in response to the user input; and

An audio renderer that

Unpacks and decompresses the audio components' subband data

10   and scale factors for each channel;

Calculates scale factors for the mixed subband data;

Mixes the audio components' subband data in the subband domain for each channel;

Compresses the mixed subband data and their scale factors for

15   each channel;

Packs and multiplexes the channels' compressed subband data and scale factors into an output frame; and

Places the output frame into a queue for transmission to a decoder.

2.   The multi-channel interactive audio system of claim 1, wherein the audio renderer mixes only the subband data that is considered audible to the user.

3.   The multi-channel interactive audio system of claim 2, wherein the audio renderer determines which subbands are audible to the user by using the listed audio components' scale factors to calculate the intra-subband masking effects and discard the inaudible audio components for each subband.

4.   The multi-channel interactive audio system of claim 3, wherein the audio renderer unpacks and decompresses the audio components' scale factors first, determines the audible subbands, and then unpacks and decompresses only the subband data in the audible subbands.

5.   The multi-channel interactive audio system of claim 4, wherein the audio renderer

a.   stores the unpacked and decompressed subband data in a left shifted format in the memory in which the sign bit of the N-bit subband data is aligned to the sign bit of the M-bit format and the M-N rightmost bits represent noise that is below a noise floor;

b.   for each subband, multiplies the audible subband data by their respective scale factor and adds them together into a sum;

c.   for each subband, multiplies the sum by the reciprocal of the maximum scale factor for the audible subband data to produce the mixed subband data;

d.   if the mixed subband data overflows the format, increments the maximum scale factor to the next largest value and repeats step c.

6.   The multi-channel interactive audio system of claim 1, wherein the input data frame further includes a header and a bit allocation table that are fixed from frame-to-frame so that only the scale factors and subband data vary.

7.   The multi-channel interactive audio system of claim 6, wherein the compressed subband data is coded with fixed length codes.

8.   The multi-channel interactive audio system of claim 7, wherein the audio renderer unpacks each piece of the N-bit subband data, where N varies across the subbands, as follows:

a.   Utilizes the FLCs and fixed bit allocation to calculate the position of the subband data in the input audio frame, extract the subband data and stores it in the memory as M-bit words where the leftmost bit is a sign bit; and

b. Left shifts the subband data until its sign bit is aligned with the M-bit word's sign bit, the rightmost M-N bits remaining in said M-bit word as noise.

9. The multi-channel interactive audio system of claim 8, wherein the audio renderer is hardcoded for the fixed header and bit allocation table so that the audio renderer only processes the scale factors and subband data to increase speed.

10. The multi-channel interactive audio system of claim 1, wherein the audio render interfaces with an application that provides equalization of the audio components, said audio renderer equalizing each said audio component by modifying its scale factors.

11. The multi-channel interactive audio system of claim 1, wherein the audio render interfaces with an application that provides for sideways localization of the audio components, said audio renderer sideways localizing the audio components by applying a phase positioning filter to the subband data that spans the range from 200Hz to 1200Hz.

12. The multi-channel interactive audio system of claim 1, wherein the input and output frames also include a header and a bit allocation table, the audio renderer providing for the seamless generation of output frames to maintain decoder sync by,

a. placing a null output template in the queue that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal;

b. if the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and

c. if the next frame is not ready, transmitting the null output template.

13. The multi-channel interactive audio system of claim 1, wherein said decoder is a Digital Surround Sound decoder that is capable of decoding multi-channel audio, said audio renderer transmitting a sequence of said output frames that provide real-time interactive multi-channel audio with the same format as the multi-channel audio.

14.   The multi-channel interactive audio system of claim 13, further comprising a single bandlimited connector, said audio renderer transmitting, in real-time and in response to the user input, the output frames as a unified and compressed bitstream over the single bandlimited connector to the Digital Surround Sound decoder, which decodes

5   the bitstream into the interactive multi-channel audio whose bandwidth exceeds that of the single bandlimited connector.

15.   The multi-channel interactive audio system of claim 1, further comprising a single bandlimited connector, said audio renderer transmitting, in real-time and in response to the user input, the output frames as a unified and compressed bitstream over the single bandlimited connector to the decoder, which decodes the bitstream into multi-channel

5   audio whose bandwidth exceeds that of the single bandlimited connector.

16.   The multi-channel interactive audio system of claim 1, wherein one or more of the audio components comprise looped data having commencing input frames and closing input frames whose subband data has been preprocessed to ensure seamless concatenation with the commencing frame.

17.   A multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames in a bitstream that is coded with fixed length codes (FLCs), each said input data frame including a header, a bit allocation table, subband data and their scale factors that

5   have been compressed and packed, said header and bit allocation table being fixed from component-to-component, channel-to-channel and frame-to-frame

A human input device (HID) for receiving input from a user;

An application programming interface (API) that generates a list of audio components in response to the user input; and

10   An audio renderer, which is hardcoded for the fixed header and bit allocation table, that

Unpacks and decompresses the audio components' scale factors for each channel;

Calculates scale factors for the mixed subband data ;

15   Uses the scale factors to determine the audible subband data;

Unpacks and decompresses only the audible subband data;

Mixes the audible subband data in the subband domain for each channel;

Compresses the mixed subband data and their scale factors for

20   each channel;

Packs and multiplexes the channels' compressed subband data and scale factors into an output frame; and

Places the output frame into a queue for transmission to a decoder.

18.   The multi-channel interactive audio system of claim 17, wherein the audio renderer unpacks each piece of the N-bit audible subband data, where N varies across the subbands, as follows:

      a.   Utilizes the FLCs and fixed bit allocation to calculate the position of the

5   audible subband data in the input audio frame, extract the audible subband data and stores it in the memory as M-bit words where the leftmost bit is a sign bit; and

      b.   Left shifts the audible subband data until its sign bit is aligned with the M-bit word's sign bit, the rightmost M-N bits remaining in said M-bit word as noise.

19.   The multi-channel interactive audio system of claim 17, wherein the decoder is a Digital Surround Sound decoder that is capable of decoding multi-channel audio.

20.   The multi-channel interactive audio system of claim 17, wherein the audio renderer generates a seamless sequence of output frames by

      a.   placing a null output template in a queue for transmission to a decoder that includes the header, the bit allocation table and subband data and scale factors that

5   represent an inaudible signal;

b.　if the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and

c.　if the next frame is not ready, transmitting the null output template.

21.　A multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames, each said input data frame including a header, a bit allocation table, and audio data that has been compressed and packed;

5　A human input device (HID) for receiving input from a user;

An application programming interface (API) that generates a list of audio components in response to the user input; and

An audio renderer that generates a seamless sequence of output frames by

a.　placing a null output template in a queue for transmission to a 10 decoder that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal;

b.　concurrently unpacking and decompressing the audio components' data and for each channel, mixing the audio components' data for each channel, calculating scale factors for the mixed data, compressing the mixed data for each 15 channel, and packing and multiplexing the channels' compressed data ;

c.　if the next frame of mixed data is ready, writing the mixed data over the previous output frame and transmitting the output frame; and

d.　if the next frame is not ready, transmitting the null output template.

22.　The multi-channel interactive audio system of claim 21, wherein the decoder is a Digital Surround Sound decoder that is capable of decoding multi-channel audio.

23.　The multi-channel interactive audio system of claim 21, wherein the audio data comprises subband data and its scale factors, the audio renderer mixing only the subband data that is considered audible to the user.

24.   The multi-channel interactive audio system of claim 23, wherein the audio renderer determines which subbands are audible to the user by using the listed audio components' scale factors to calculate the intra-subband masking effects and discard the inaudible audio components for each subband.

25.   The multi-channel interactive audio system of claim 24, wherein the audio renderer unpacks and decompresses the audio components' scale factors first, determines the audible subbands, and then unpacks and decompresses only the subband data in the audible subbands.

26.   A multi-channel interactive audio system, comprising:

A memory for storing a plurality of audio components as sequences of input data frames, each said input data frame including a header, a bit allocation table, subband data and their scale factors that have been compressed and packed;

5   A human input device (HID) for receiving input from a user;

An application programming interface (API) that generates a list of audio components in response to the user input and calculates mapping coefficients that map each audio component on the list to each channel in a digital surround sound environment;

10   An audio renderer that

Unpacks and decompresses the audio components' subband data and scale factors for each channel;

Calculates scale factors for the mixed subband data;

Mixes the audio components' subband data in the subband domain

15   for each channel;

Compresses the mixed subband data and their scale factors for each channel;

Packs and multiplexes the channels' compressed subband data and scale factors into an output frame; and

20   Places the output frame into a queue; and

A Digital Surround Sound decoder that decodes the output frames to produce multi-channel audio, said output frames having the same format as existing prerecorded multi-channel digital audio.

27.   A multi-channel interactive audio system, comprising:

A human input device (HID) for receiving input from a user;

A console, comprising:

A memory for storing a plurality of audio components as sequences of

5   input data frames, each said input data frame including subband data and their scale factors that have been compressed and packed;

An application programming interface (API) that generates a list of audio components in response to the user input; and

An audio renderer that

10   Unpacks and decompresses the audio components' subband data and scale factors for each channel;

Calculates scale factors for the mixed subband data;

Mixes the audio components' subband data in the subband domain for each channel;

15   Compresses the mixed subband data and their scale factors for each channel;

Packs and multiplexes the channels' compressed subband data and scale factors into an output frame; and

Places the output frame into a queue where the compressed audio

20   data is output as a seamless unified bitstream;

A digital decoder that decodes the bitstream into a multi-channel audio signal; and

A single bandlimited connector that delivers the bitstream to the decoder.

28.   A method of preparing PCM audio data for storage in a compressed format that is compatible with looping, wherein said PCM audio data is stored in a file and the compressed format includes a sequence of compress audio frames comprising:

a. Compacting or dilating the PCM audio data in time to fit the boundaries

5 defined by a whole number of compressed audio frames to form a looped segment;

b. Appending N frames of PCM audio data from the end of the file to the start of the looped segment;

c. Encoding the looped segment into a bitstream,

d. Deleting N compressed frames from the beginning of the encoded

10 bitstream to yield a compressed audio loop sequence in which the compressed audio data in the closing frames of the loop sequence ensure seamless concatenation with the commencing frames during looping.

29. A method of rendering multi-channel audio, comprising:

a. Storing a plurality of audio components as sequence of input data frames, each said input data frame including subband data and their scale factors that have been compressed and packed;

5 b. In response to a user input, generating a list of audio components;

c. Unpacking and decompressing the subband data and scale factors for each channel;

d. Calculating scale factors for the mixed subband data;

e. Mixing the subband data for each channel;

10 f. Compressing the mixed subband data and their scale factors;

g. Packing and multiplexing the channels' compressed subband data and scale factors into an output frame; and

h. Placing the output frame into a queue for transmission to a decoder.

30. The method of claim 29, wherein unpacking and decompressing the subband data comprises:

unpacking and decompressing only the scale factors;

using the scale factors to determine which subbands are audible;

5 unpacking and decompressing only the audible subband data.

31.　　The method of claim 30, further comprising sideways localizing the audio components by applying a phase positioning filter to the subband data that span the range from approximately 200Hz to approximately 1200Hz.

32.　　The method of claim 29, further comprising:

　　　a.　　placing a null output template in a queue for transmission to a decoder that includes the header, the bit allocation table and subband data and scale factors that represent an inaudible signal;

5　　　b.　　if the next frame of mixed subband data and scale factors is ready, writing the mixed subband data and scale factors over the previous output frame and transmitting the output frame; and

　　　c.　　if the next frame is not ready, transmitting the null output template.